Rainbow1 IP Phone User Manual

Version: 1.0

2014-03-27

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Contact ATCOM

Overview of ATCOM

ATCOM is the leading VoIP hardware manufacturer in global market. We have been keeping innovating with customer's needs oriented , working with partners to establish a total solution for SMB VoIP with IP phone , IP PBX and Asterisk cards

With over 10 years' experience of R&D , manufacturing and service in network and VoIP filed ; mission of creating the biggest value for IP terminals , we commit ourselves in supplying the competitive IP phone and other terminals for IP PBX , softswitch , IMS , NGN providers and carriers; supplying the competitive total VoIP solution for SMB market. We keep improving the customer's experience and creating the bigger value with our reliable products. Until now, our VoIP products have been available in 100+ countries and used by millions of end users.

Contact Sales

Address	Area C, A2F , Block 3 ,Huangguan Technology Park , #21 Tairan 9th Rd, Chegongmiao , Futian District , Shenzhen China
Tel	+ (86) 755-83018618-8806
Fax	+ (86) 755-83018319
E-mail	sales@atcomemail.com

Contact Technical Support

Tel	+ (86) 755-83018618-8008
E-mail	Support@atcomemail.com

Website Address: http://www.atcom.cn/

Download Center: http://www.atcom.cn/download.html

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1. Overview of Rainbow 1

Rainbow1

Type	POE	Power adapter
Rainbow1	Yes	Optional accessory

1.1 Interfaces

- Power input: DC 5V, 1000mA or POE
- LAN: RJ45 port
- PC: RJ45 port
- Headset jack 1 : RJ9 port
- Handset jack 1 : RJ9 port

1.2 Hardware

- LCD: 132×52
- FLASH: 16M
- RAM: 16M
- CPU: 262MHz Dual Core
- LED indicator: 1 Status Light

1.3 Software

- Sip 2.0 (RFC3261) and other related SIP RFCs
- 1 SIP lines
- STUN
- Jitter Buffer, VAD,CNG

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- G.711A/u, G722, G.723, G.726-16, G.726-24, G.726-32, G.726-40, G.729 , Lin16-16, iLBC
- Echo Cancellation
- SIP Domain name, Authentication
- DTMF（inband, RFC2833, info）
- Call transfer, Call forward, 3-way conference, Call hold, Call back
- DND(Do Not Disturb), Auto answer, Blacklists, Block Call-ID, Block Anonymous call, Dial plan, IP call
- Phone book with 100 white records and 50 black records, 100 answered calls, 100 missed calls, 100 dialed calls
- Update via HTTP, FTP, TFTP, PNP
- Syslog
- SNTP
- WEB access with different login level
- Multi-language: English, Chinese, Farsi, French, German, Hebrew, Italian, Portuguese, Russian, Spanish, Turkish
- Soft button: soft button * 3
- Redundancy SIP server

1.4 Network

- LAN/PC：Support bridge mode
- Support PPPoE（ADSL,cable modem used for Internet connecting）
- Support VLAN（DATA VLAN and VOICE VLAN）
- Support L2TP VPN
- LAN support Primary and Secondary DNS
- LAN support DHCP Client
- Support QoS

1.5 Management and Maintenance

- Support safe mode and firmware updating under safe mode
- Support different level user management
- Configuration via web , keyboard
- Support multi-language
- Auto provision (Firmware and configuration file)
- Support system log and call log

1.6 Protocol

- IEEE 802.3 /802.3 u 10 Base T / 100Base TX
- PPPoE: PPP over Ethernet
- DHCP: Dynamic Host Configuration Protocol
- SIP RFC3261, RFC3262, RFC3263, RFC3264, RFC3265, RFC2543, RFC3489, RFC3842, RFC3515, RFC2976, RFC3428, RFC2327, RFC2782, RFC1889
- TCP/IP: Transfer Control Protocol/Internet Protocol
- RTP: Real-time Transport Protocol
- RTCP:RTP Control Protocol
- DNS: Domain Name Server
- TFTP: Trivial File Transfer Protocol
- HTTP:Hypertext Transfer Protocol
- FTP:File Transfer Protocol

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1.7 Compliant and Authenticated Standard

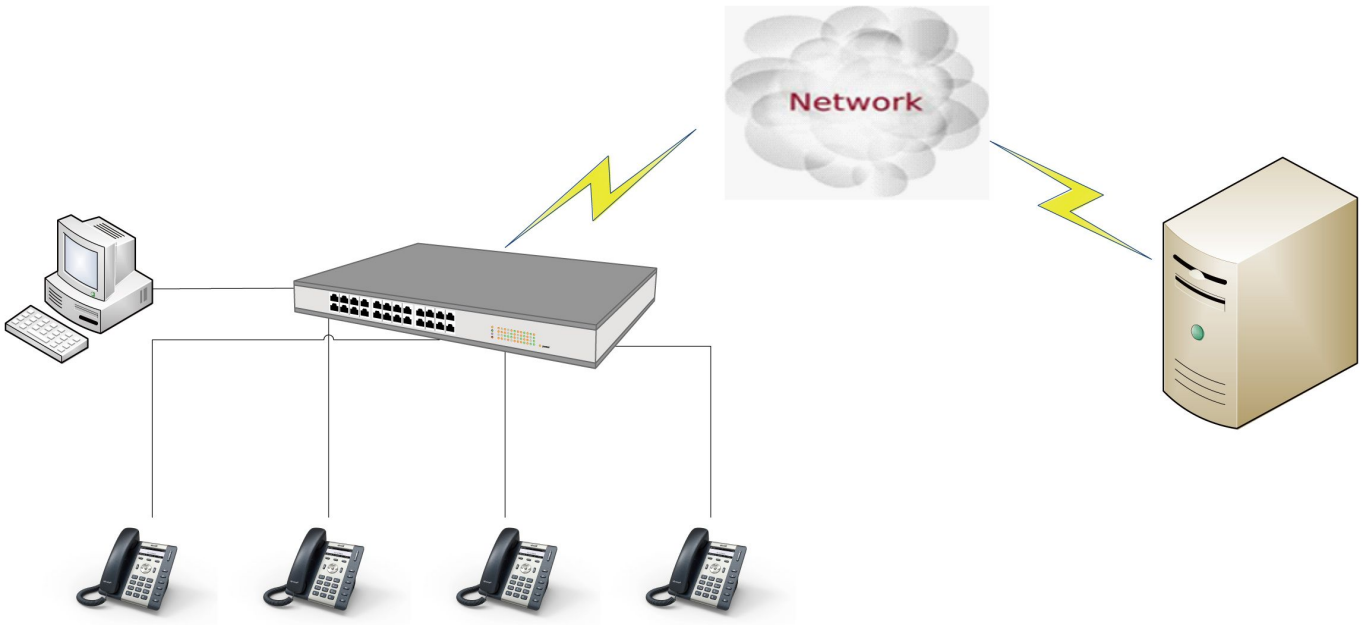
- CE: AGC01180140201E2，AGC01180140202E2
- Comply with ROHS in EU
- Comply with ROHS in China
-
- Operation temperature: lower than 60° C
- Storage temperature: lower than 60° C
- Humidity: 10 to 90% no dew

1.8 Packing List

Model	Phone	Handset	Handset line	Stand bracket	Ethernet Cable	User Manual CD
Rainbow1	1	1	1	1	1pcs 1.5m	1

Note: Power adapter (Input: AC 100~240V, 50/60Hz; Output: 5V, 1000mA;)

1.9 Installation



Connect LAN port to PC with Ethernet cable, or connect Rainbow1 to a switch/router which is in the same network as your PC. It uses DHCP mode by default, and you can review its current IP address by pressing key on idle state. To access the web interface, you can input the IP address in IE browser. E.G. The IP address of your Rainbow1 is 192.168.1.100, you can input 192.168.1.100 and press enter key on your browser to access its webpage. There are two login level:

User

Admin

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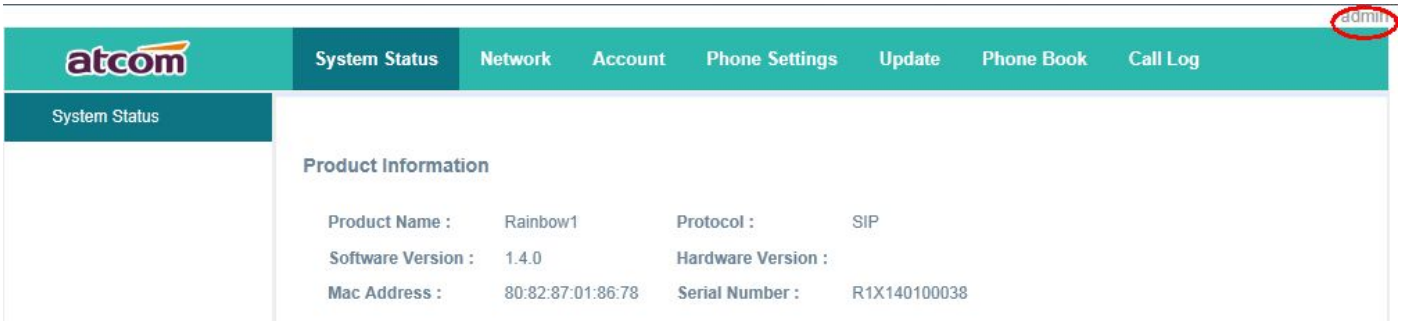
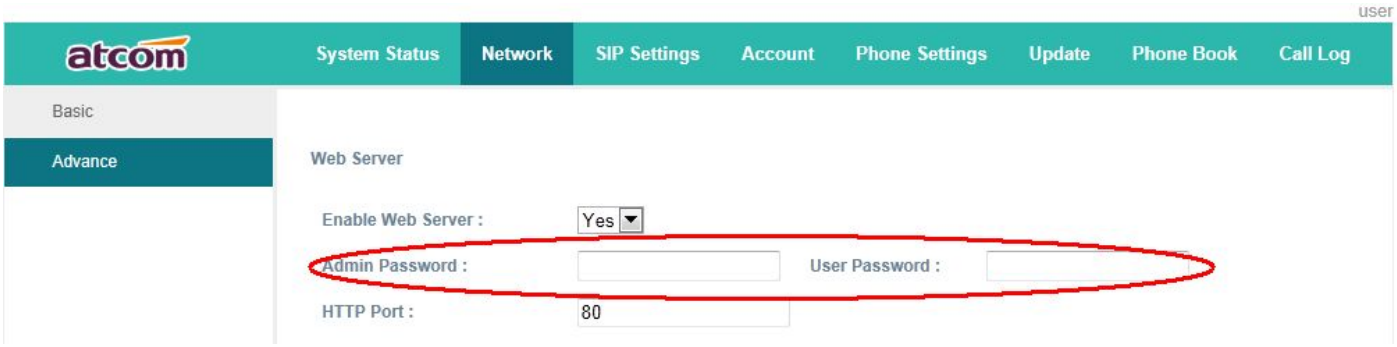


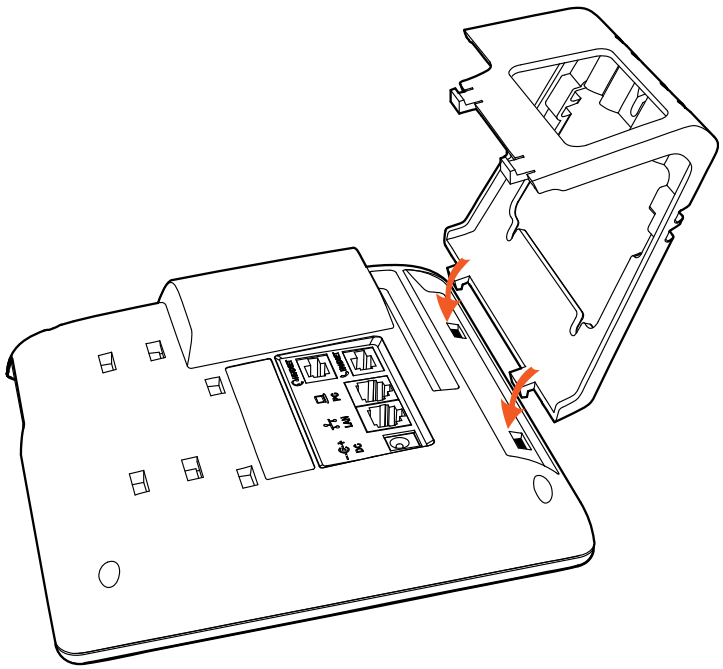
No password is set for those two accounts with factory settings. You can click admin button on the right corner of the webpage to switch from the user to admin mode. To set the password for user and admin login you can firstly login as admin and enter the Network--->Advance page as following.



Installation instruction

1. Desktop installation

A. Put the bottom side of the IP phone upside and press one-side joints of stand bracket into the slot, please refer the picture as below:



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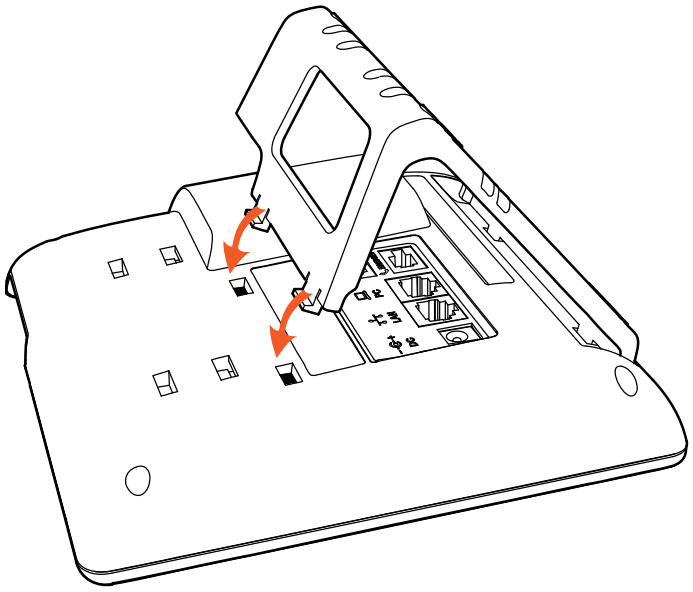
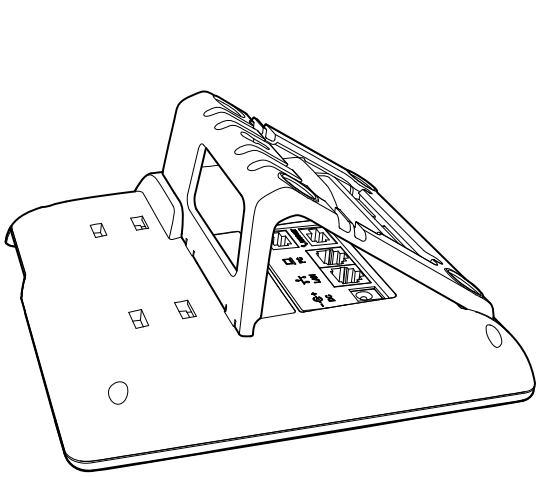
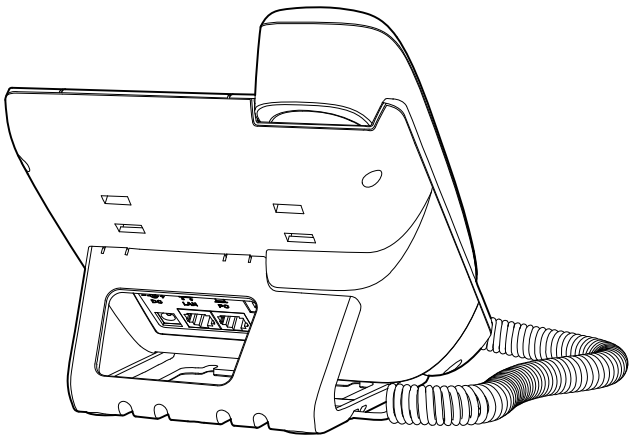
B. Press the other side joints into the slot according to the direction of the arrow



C. It is the right picture after fixing the stand bracket below:





D. Disassemble the stand bracket:

Push the spring joint of stand and pull the stand according to the direction of the arrow. When the joints are pulled out of the slot, you can take off the stand bracket.

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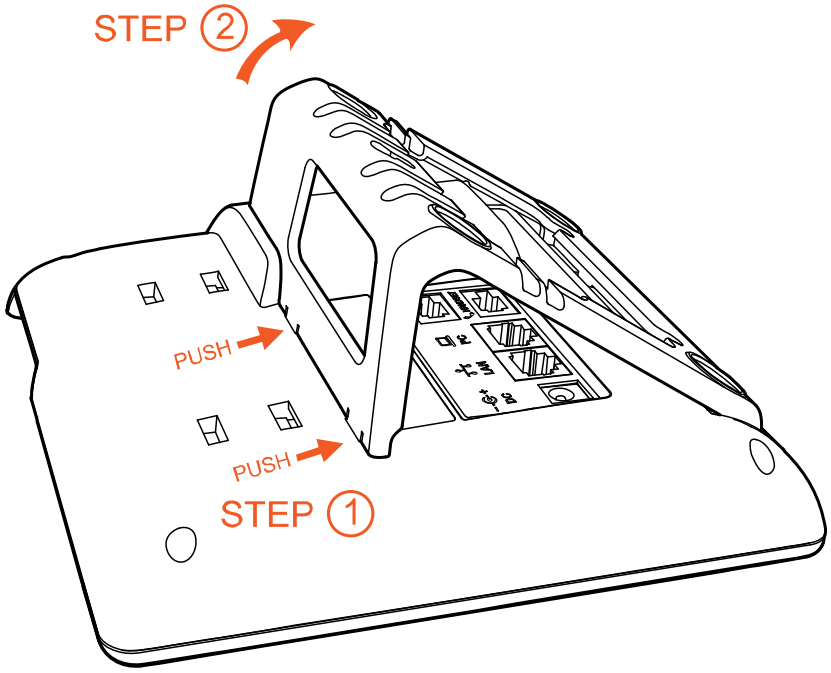
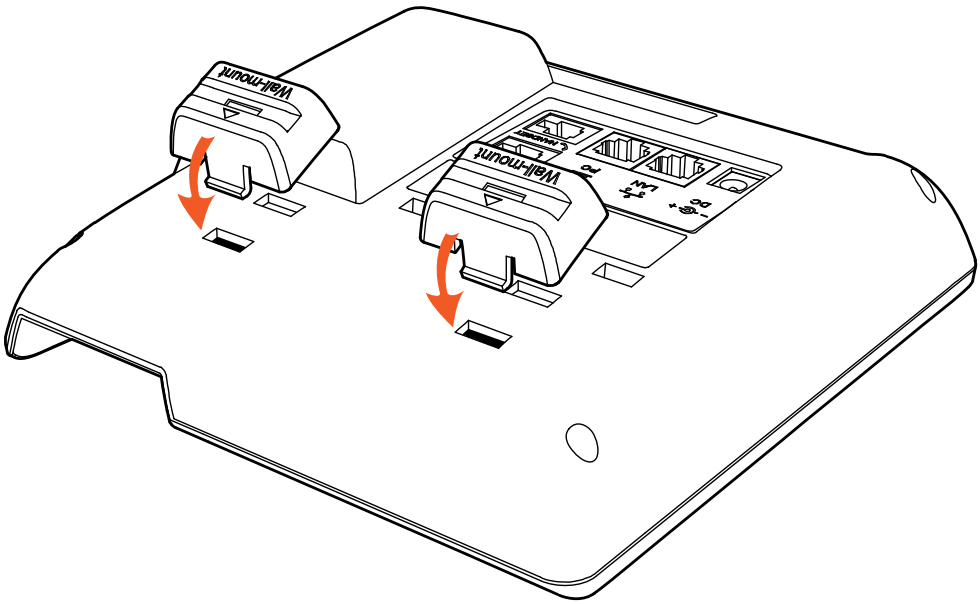


2. Wall-hung Installation

A. Put the bottom side of the IP phone upside and press one-side joints of wall-hung stand bracket into the slot, please refer the picture as below:



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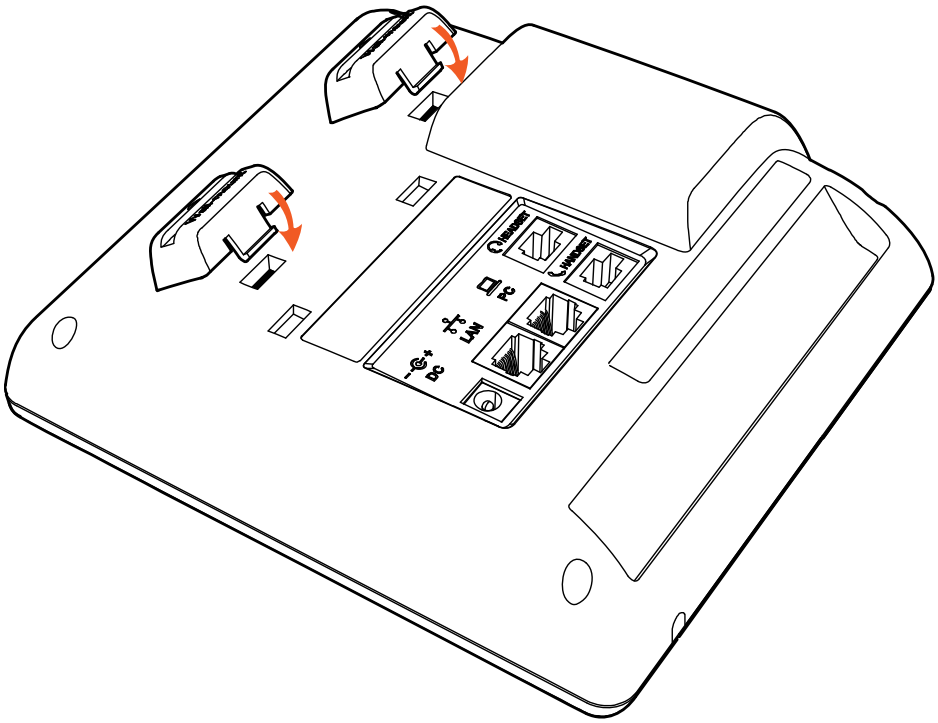
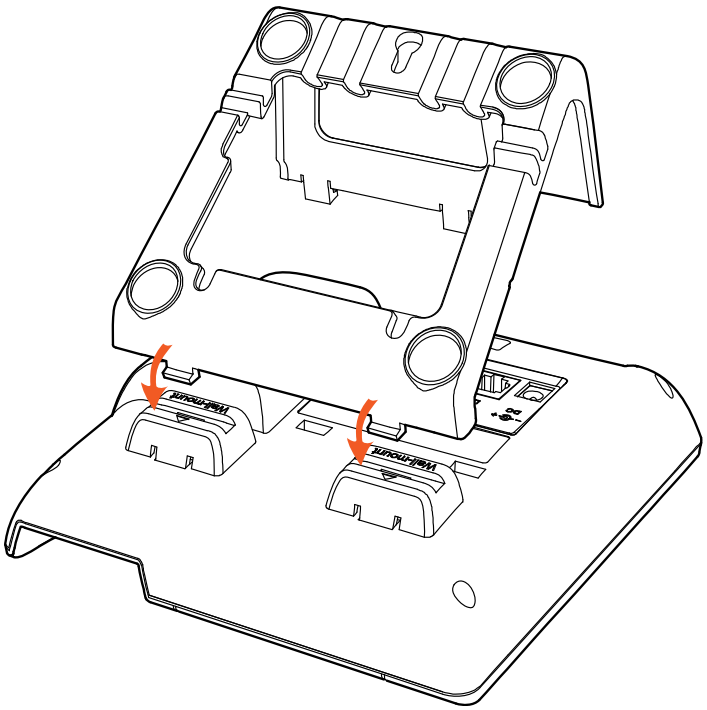
B. Press the other side joints into the slot according to the direction of the arrow



C. After install the wall-hung stand bracket, press one-side joints of stand bracket into the slot, please refer the picture as below:



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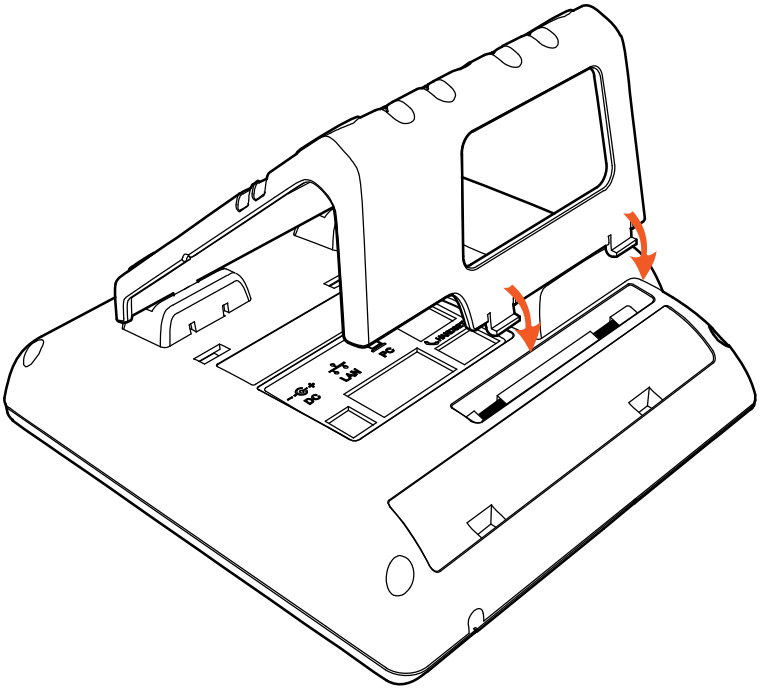
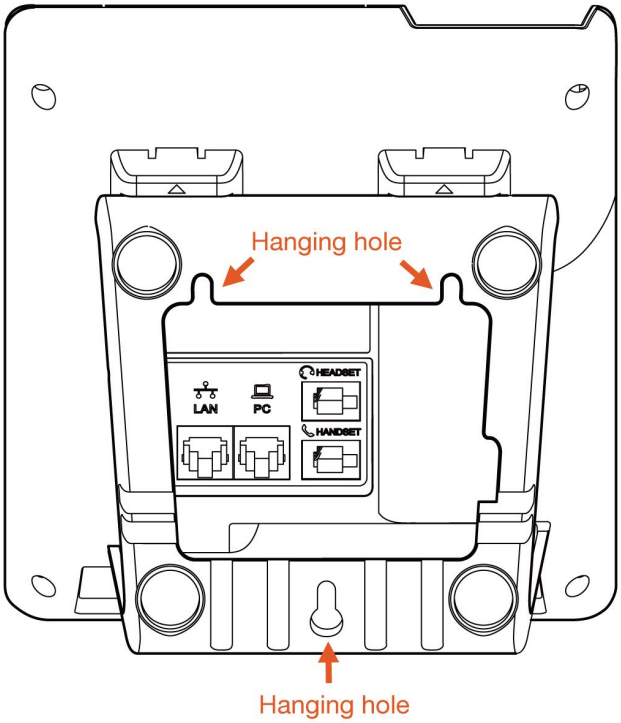
D. Press the other side joints into the slot according to the direction of the arrow



E. Knock in nails or screws on the wall according to the proportion of the distance between the hanging holes as below:



F. It is the right picture after fixing the stand bracket below:

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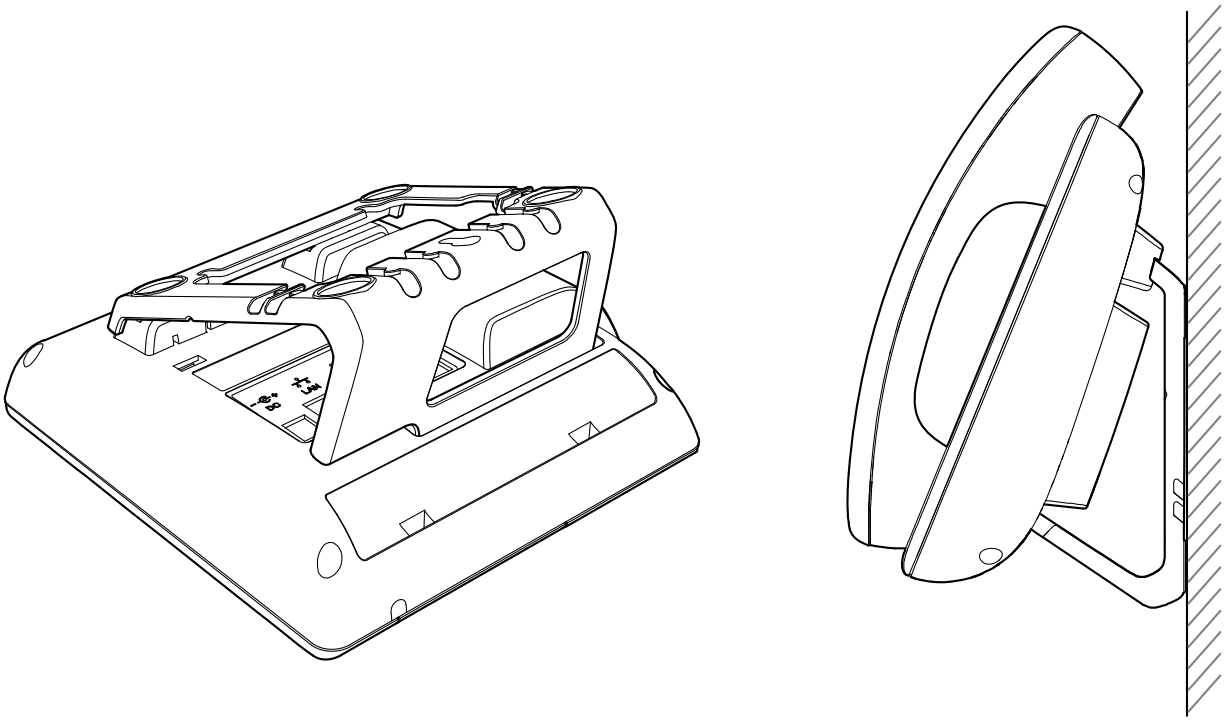
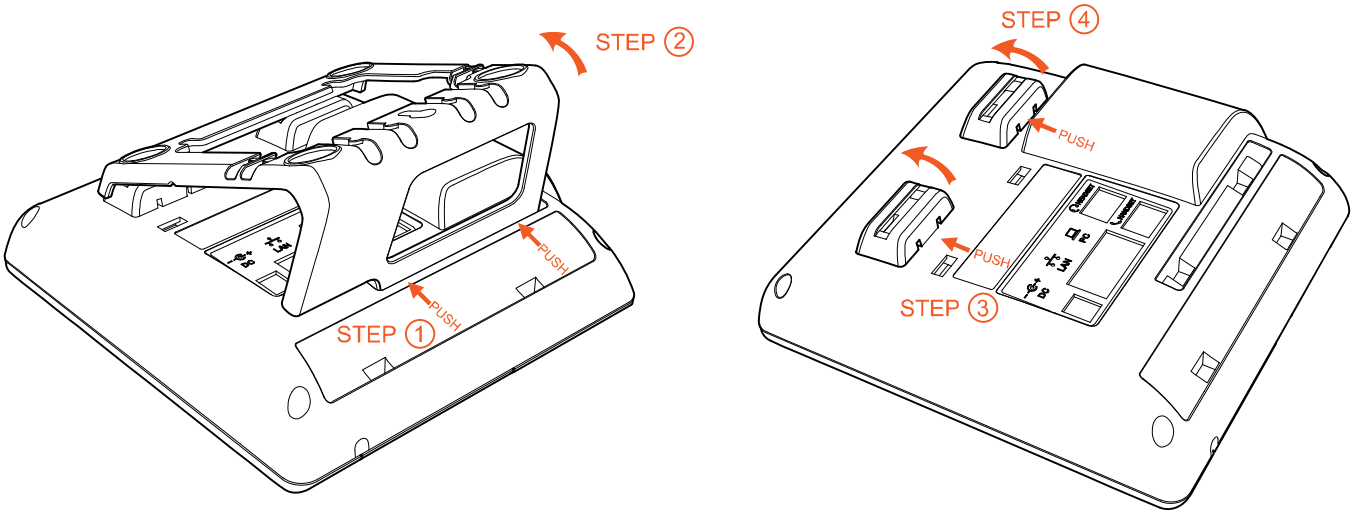


G. Disassemble the stand bracket:

Push the spring joint of stand and pull the stand according to the direction of the arrow. According to the direction of the arrow:



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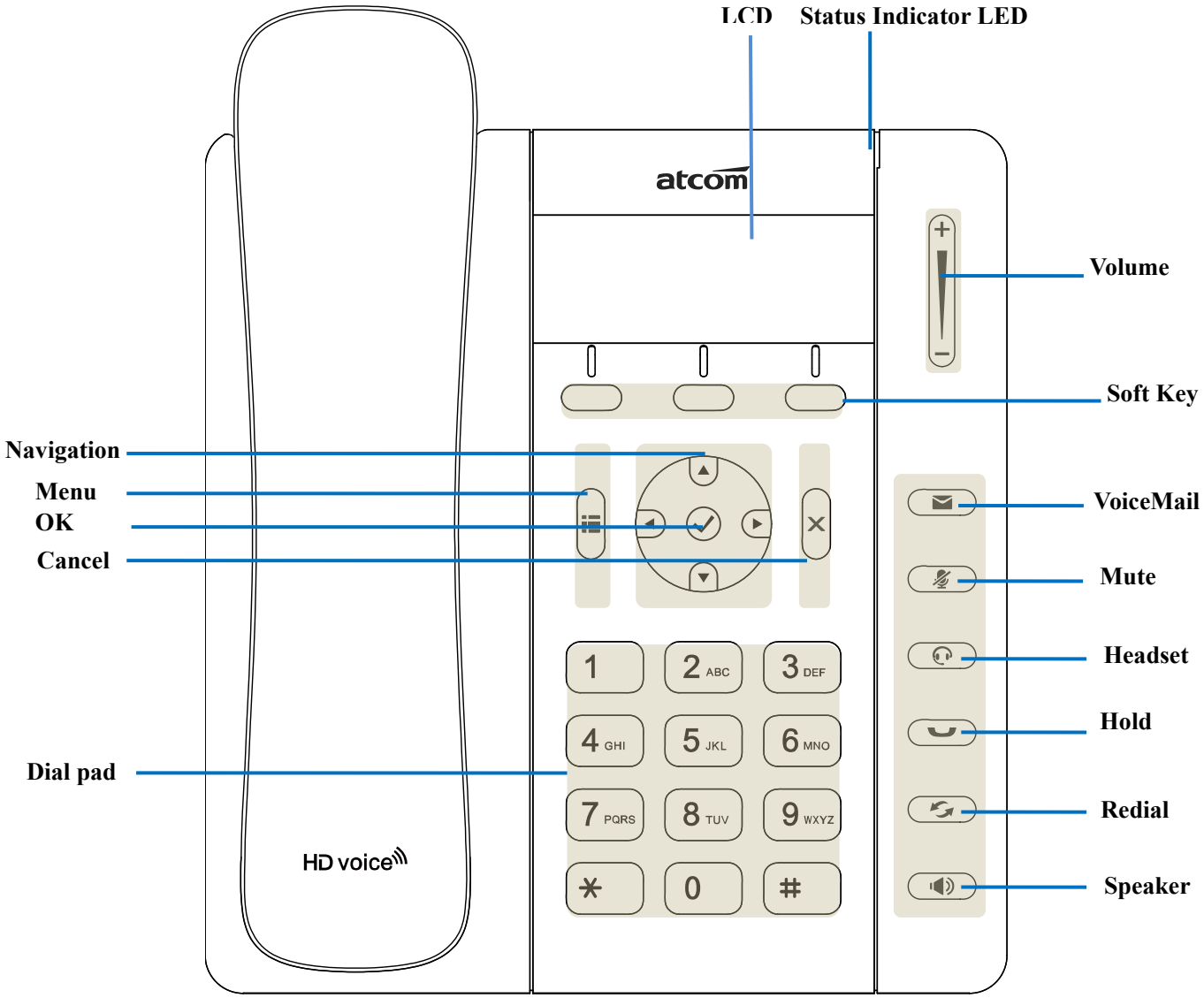
2. Keypad of IP Phone



2.1 Describe of the buttons and Screen

Soft Keys	Press to select a function which displayed at the softkey zone of screen Display all optional functions at the last line of LCD screen.
Status Indicator LED	Indicates the phone's status, ➢ If the phone is being started ,the LED is on ➢ If the phone is standby, the LED is off ➢ If there is income calling, the LED will blink at 120ms on, 120ms off. ➢ If there is a new message, the LED will blink at 600ms on, 600ms off.
LCD Screen	All information such as date, time, phone number, incoming caller's ID(if available),line/call status, extension numbers and the soft key features are displayed on it.

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Dial pad	Entering numbers or characters.
Navigation keys	Allows users to navigate (left, right, up, down). Press ▼ can enter "Directory". Press ▲ can enter "History".
	Confirm users' operation and show status when Rainbow1 is idle.
	Cancel users' operation.
	Enter menu settings.
	Adjust the volume (speaker/handset/headset/ring). Volume level will be displayed on LCD when pressing volume key.
	Pick up and hung up under the speaker mode.
	Mute the mic in a conversation by pressing the Mute button; this prevents the person on the active call from hearing what you or someone else in the room is saying. To un-mute, press the Mute button again.
	Pick up and hung up under headset mode.
	Check the Voicemail status.
	Hold the current call.
	Redial the last dialed number

2.2 Describe of the LCD Screen Icons

Icon	Description
	The extension is registered
	The extension is unregistered
	There is a new voice mail
	There is an incoming call

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	The call is held
	Rainbow1 is in speaker mode
	Rainbow1 is in handset mode
	Rainbow1 is in headset mode
	The call is muted

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3. Basic functions and operations

3.1 Answer the calls

When there is an incoming call, phone will remind user with ringing. There are 3 ways to answer the call

A. Answer by handset

Pick up the handset and talk with the caller. If you want to hang up, just put back the handset. When you are talking with the handset and want to switch to speaker mode or headset mode, please press key or key, then put down the handset.

B. Answer by speaker

Press key and talk with callers by built-in Micro-phone and Speaker. If you want to hang up, please press key again. Switch calling or talking into handset mode by lifting the handset under speaker mode. Press key will switch calling or talking into headset mode.

C. Answer by headset

Keep your microphone connected with the RJ9 headset jack, when there is an incoming call, press key and talk with the caller. If you want to hang up, please press key again. Pressing key can change calling or talking into speaker mode, and lifting the handset switches to handset mode.

3.2 Make Call

A. Use the handset

Pickup the handset and input a phone number. Press soft key "Send" to dial the number. When you hear the tones of "du~~du~~" and the phone number your dialed is being displayed on the LCD, the phone at the side of being called should be ringing. If the called party answers this calling, the call is established and the calling timer is started immediately.

B. Use the speaker

Press key and input a phone number. Press soft key "Send" to dial the number. When caller hear the tones of "du~~du~~" and the phone number your dialed is being displayed on the LCD, the phone at the side of being called should be ringing. If the called party answers this calling, the call is established and the calling timer is started immediately.

C. Use the headset

Press Key and input a phone number. Press soft key "Send" to dial the number. When caller hear the tones of "du~~du~~" and the phone number your dialed is being displayed on the

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LCD, the phone at the side of being called should be ringing. If the called party answers this calling, the call is established and the calling timer is started immediately.

D. Dial from phone book

1. Press key and input the keypad password 123 to enter the menu and choose "Directory" option. Press "Select" soft key and then find the contact person by navigation keys. When the certain contact person is highlighted, press "Dial" or just pick up the handset to call this number.
2. Pick up the handset, press "Directory" soft key, then select the contact person and press "Dial" soft key.
3. Pick up the handset, press and enter "Directory", then select the contact person and press "Dial" soft key.

E. Dial from call history

1. Press key and input the keypad password 123 to enter the menu and choose "History" option, then enter sub-directory "Dialed Calls", "Received Calls" or " Missed Calls" to select one of call history entry, and press "Dial" soft key or pick up handset to call this number.
2. Pickup the handset, press "History" soft key, then select one of call history entry, and press "Dial" soft key to call this number.
3. Pickup the handset, press and enter "Call History" ,then select one of call history entry, and press "Dial" soft key to call this number.

3.3 Pre- dial

It's a method to dial a phone number immediacy at standby mode.

A、Dial-up the phone number at standby mode

B、Press soft key "Dial" to send out the number

3.4 Multiple line dial-up

Rainbow1 supports up to 2 concurrent calls. If there is a new incoming call when you're talking on Rainbow1, the new incoming call will be displayed on LCD and status indicator LED will be fast blinking. User can press soft key "Hold" then press "Answer" to receive the new incoming call.

3.5 Hang up the phone

1. Handset hang up

Put back the handset at handset mode, the current calling will be hung up.

2. Speaker hang up

Press key at speaker model, the current calling will be hung up.

3. Headset Hang up

Press key at headset model, the current calling will be hung up.

4. Hang up one line call

Press the hook to hang up the current calling when 2 calls happened simultaneously.

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3.6 Call Transfer

1. Attended call transfer

The attended transfer allows user to call a third-party before transferring the calling.

While on calling, press the "Transfer" soft key to hold the current call and phone the third party. Then dial the target number you want to transfer to on the activated line and press “Send” soft key to call that number. After the target party answers the call, press "Transfer" soft key again to complete the transfer.

2. Blind call transfer

The blind transfer allows user to transfer a call without speaking to the third party. On the user side, the call will be ended as soon as the target phone number is dialed.
Operating steps: Press “More” soft key to get more option, then press “Bxfer” soft key, input the transfer target number and press “Send” soft key.

3.7 Voicemail

Rainbow1 has a key for entering voicemail box. Press key to enter the menu to configure voicemail number if never configure it previously. Otherwise, the voicemail number will be called after press it. If you want to modify it after configured it, please go to the Account webpage to modify the voicemail number.

3.8 Mute calls

The input audio will be not transmitted to peer phone after pressing key, and the phone will be muted even switched among different modes of speaker, handset and headset. To un-mute, just press key again.

3.9 Call Hold

The current calling will be hold by pressing soft key “Hold” or key. And the held call will be resumed after pressing soft key “Resume” or key or the corresponding line key. Even on 3-way conference calling, the 3-way conference calling will be held after pressing “Hold” key, and be resumed to 3-way conference after pressing “Hold” Key again. Remember the conversation is still on hold without being ended even if hung up under the status of hold.

3.10 3-Way Conference Call

To initiate a conference call:
Press “More” soft key on calling to get more options, then press “Conf” soft key to start a conference call. Enter the phone number of the third party and press “Send” soft key to send it out.

1. After the third party answers the call, pressing "Conf" key again to establish the 3-way conference.
2. 3-way conference initiator can press "Exit" soft key to quit from the conference and leave the other two parties still in the conversation.
3. If the initiator hangs up the call or press the “End Conf” soft key, the conference will be ended and the calling between the other two parties will be hung up.

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3.11 Call History

Rainbow1 supports 100 missed calls list, 100 incoming calls list and 100 dialed calls list. When the storage is full, the old record will be erased by the new one.

Press "History" soft key or key when Rainbow1 is standby, all the incoming(->), outgoing(<-) and missed calls(!) will be listed. There is other ways to check them:

A. Missed call

1. Press key.
2. Press key and key to select "Call History" then press "Select" soft key.
3. Press key and key to select "Missed Calls" then press "Select" soft key.
4. Press key and key to browse the missed call record. If there is no record, the LCD display will be indicated "List is Empty".

B. Answered call

1. Press key .
2. Press key and key to choose "Call History" and then press "Select" soft key.
3. Press key and key to choose "Answered Calls" and then press "Select" soft key.
4. Press key and key to browse the answered call records. If there is no record, the LCD display will be indicated "List is Empty".

C. Dialed call

1) Press key.
2) Press key and key to select "Call History" and then press "Select" soft key.
3) Press key and key to select "Dialed Calls" and then press "OK" soft key.
4) Press key and key to browse the dialed call records. If there is no record, the LCD display will be indicated "List is Empty".

3.12 Default Password

Password is needed to access menu and web.

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The default password for accessing menu is 123.

There are 2 modes to access web: User mode and Admin mode. All the parameters can be visible at Admin mode while part of them can be visible at User mode.

- User Mode
  Username：user no default Password
- Admin Mode：
  Username：admin no default Password

3.13 Check the Phone's IP address

Press key, then the status of the phone will be displayed on the screen and you will see the current IP address of the phone.

3.14 Dial Plan

Dial plan defines the rule of dial. The syntax of dial plan for Rainbow1 is closely similar as the corresponding syntax specified by MGCP and MEGACO.

Dial plan is stipulated by the below configurable parameters:

- Interdigit Long Timer——refer to Interdigit long timer for more detail description
- Interdigit Short Timer——refer to Interdigit short timer for more detail description
- Dial Plan

Dial plan contains a series of digit sequences, separated by the '|' character. The collection of sequences is enclosed in parentheses '(' and ')'.

Default: (*xx.|xxxxxxxxxxx.)

When user dials a series of digits, Rainbow1 will response in below way:

- No candidate sequences matched, the number will be rejected and "call ended" will be displayed on the screen. For instance, if the default dial plan only supports digits, any '*' character or letters input will be rejected.
- More than one candidate sequences matched, Rainbow1 will wait for more digits input.
- When input timeout occurs, Rainbow1 will dial the digits input already.
- When input '#' character, Rainbow1 will dial the input digits immediately.

Digit Sequence Syntax:

'x': Matches any one numeric digit ('0' ... '9')

'[]': Numeric ranges are allowed within '[]'. For example, [389] means '3' or '8' or '9', [3-6] means '3' or '4' or '5' or '6' , [235-8*] means '2' or '3' or '5' or '6' or '7' or '8' or '*'.

'.' : Any element can be repeated zero or more times by appending a '.' character. For example, 01. matches 01,011,0111,01111,...,011111111...etc.

'<>': subsequence substitution. For example, '<8:1650>xxxxxxx' would match '85551212' and the first digit '8' will be replaced by '1650' and '16505551212' will be dialed.

',': An "outside line" dial tone can be generated within a sequence by appending a ',' character between digits. Thus, the sequence "9, 1xxxxxxxxxx" sounds an "outside line" dial tone after the user presses '9', until the '1' is pressed.

'!': A sequence can be barred (rejected) by placing a '!' character at the end of the sequence. For example, "137xxxxxxxx!" will forbid numbers which have 11 digits and start by 137 to be dialed.

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Example:

(xxxxxxx|[*#]xxxx|9,1xxxxxxxxxx|00xxx!) contains 4 subsequences:

1. Allow to dial numbers with 7 digits
2. Allow to dial numbers with 4 digits and start by '*' or '#'
3. Allow to play an "outside line" dial tone after pressing '9' and dial numbers with 11 digits and start by 1
4. Forbid to dial numbers with 5 digits and start by 00

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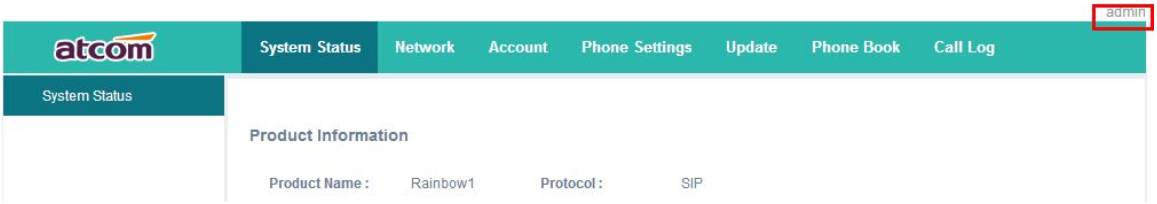
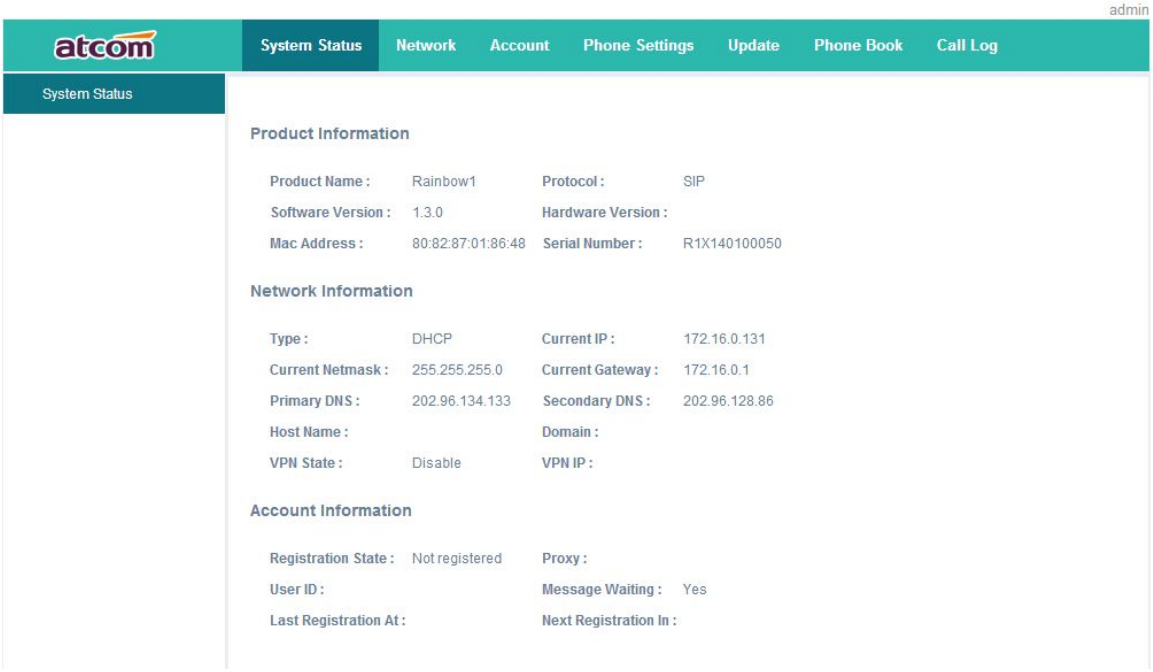
4. Web settings

Input the IP address in the web browser and press ‘Enter’ key to access Rainbow1's user webpage. Click "admin" which is on the right top corner to enter administrator webpage.



4.1 System status



This page shows the IP phone's system status.

1. **Product Information** shows the product name, protocol, software and hardware version, Mac address and serial number.

2. **Network Information** shows the network connection type, IP address, netmask, gateway, DNS, host name, domain and VPN information.

3. **Account Information** shows registration state, proxy, user ID, message waiting on or off , the last and next registration time.

4.2 Network

4.2.1 Basic

There are 3 ways to connect to the internet: DHCP, Static and PPPoE, please choose one according to your own situation.

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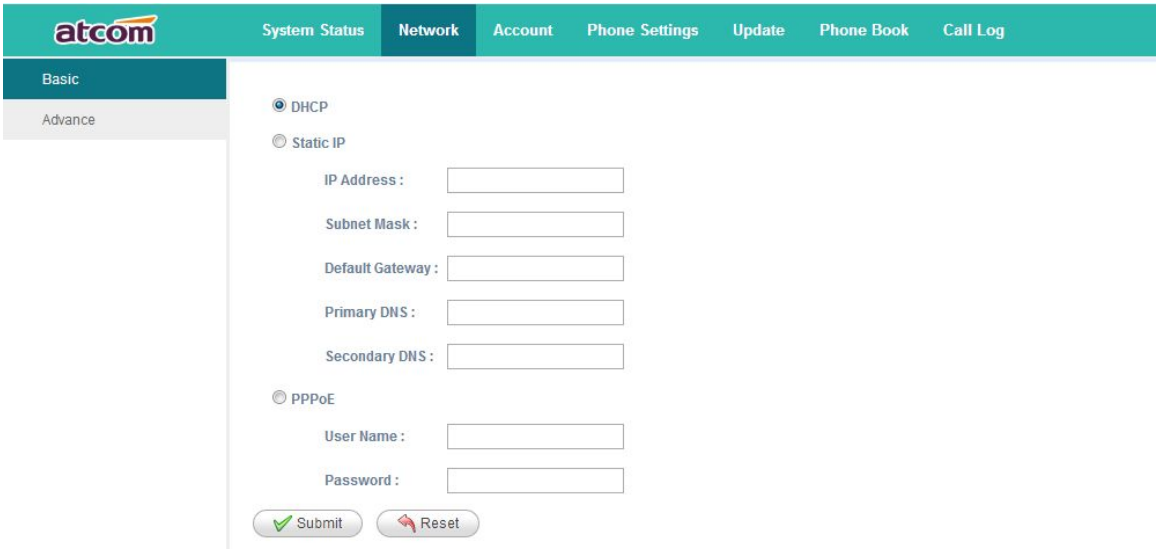


Network mode: DHCP, Static IP and PPPoE

1. DHCP

   Obtain automatically dynamic IP from DHCP server.

2. Static IP

   a) IP Address: set IP address

   b) Subnet Mask：set netmask

   c) Default Gateway：set gateway

   d) Primary DNS：set primary DNS server

   e) Secondary DNS：set secondary DNS server

3. PPPoE

   Connect the LAN port of Rainbow1 with the Ethernet port of modem then fill user name and password of PPPoE which assigned by ISP (Internet Service Provider).

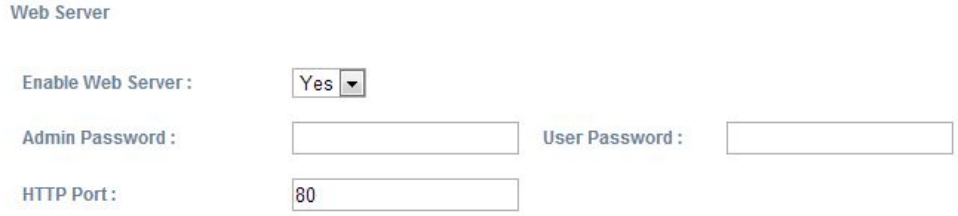
a) User name： set PPPoE user name

b) Password：set PPPoE password

Press 'Submit' button after finishing setting and all the settings info will be saved and taken effect after Rainbow1 reboots.

4.2.2 Advance

➢ **Web Server**



✓ Enable Web Server: Enable or disable web access. If choose "no", you're not able to access

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Rainbow1's webpage.

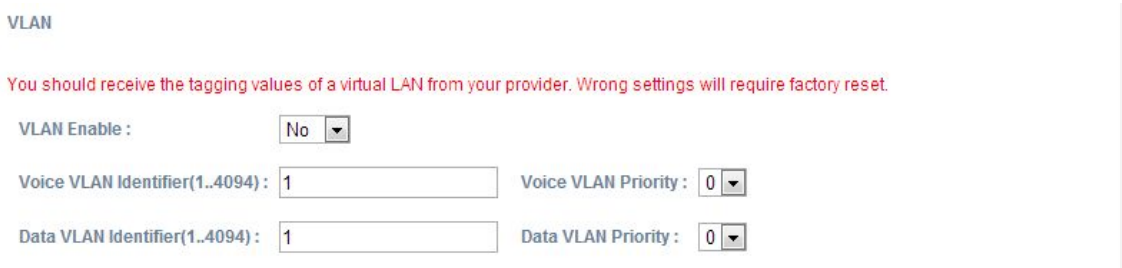
- ✓ Admin password: Set password for admin webpage access. Input 'http://ip-address/index.asp' in the web browser to access admin's login webpage after setting the admin password, then input username(admin) and password to access the admin's webpage.
- ✓ User password: Set password for user webpage access. Input 'http://ip-address/user.asp' in the web browser to access user's login webpage after setting the user password, then input username(user) and password to access the user's webpage.
- ✓ HTTP port: set port for HTTP access (defaults to 80)
  For example, Rainbow1's IP is 192.168.1.223
  HTTP port was set as 100, you have to type "http://192.168.1.223:100" in web browser to enter Rainbow1's webpage.

❖ VPN

1. VPN Enable: Select "Yes" or "No" to enable or disable VPN
2. VPN Type: Only support L2TP VPN
3. Server address: VPN server address
4. Username: VPN account's username
5. Password: VPN account's password

After apply, the phone will be reboot. The VPN IP address will be shown on the System Status webpage.

❖ VLAN



1. VLAN Enable: Select "Yes" or "No" to enable or disable VLAN
2. Voice VLAN Identifier(1..4094) : Assign VLAN ID for voice stream, range from 1 to 4094
3. Voice VLAN Priority: range from 0 to 7, 7 is the highest priority.
4. Data VLAN Identifier(1..4094) : Assign VLAN ID for data stream, range from 1 to 4094
5. Data VLAN Priority: range from 0 to 7, 7 is the highest priority.

➢ Port Link

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Choose the network type and port link of LAN/PC

1. LAN Port Link: Auto negotiate, full duplex 10Mbps, full duplex 100Mbps, half duplex 10Mbps, half duplex 100Mbps.
2. PC Port Link: Auto negotiate, full duplex 10Mbps, full duplex 100Mbps, half duplex 10Mbps, half duplex 100Mbps.

➢ Qos

QoS: Quality of service

1. SIP Qos: Quality of service for SIP (Diffserv)
2. Voice Qos: Quality of service for RTP (Diffserv)

➢ Syslog

1. Enable Syslog: Select "Yes" or "No" to enable or disable syslog.
2. Log level: None，Alert，Critical，Error，Warning，Notice，Info，Debug. The debug level is the most detailed.
3. Syslog Server: Syslog Server address.
4. Port: Syslog server port, defaults to 514.

4.3 SIP Setting

➢ Sip Timer Values

1. Sip T1: RFC 3261 T1 value (RTT). Range: 0 – 64 sec, defaults to 0.5
2. Sip T2: RFC 3261 T2 value (Maximum retransmit interval for non-INVITE requests and INVITE responses). Range: 0 – 64 sec, defaults to 4
3. Sip T4: RFC 3261 T4 value (Maximum duration a message will remain in the network). Range: 0 – 64 sec, defaults to 5
4. Reg Retry Intvl: Interval to wait before the phone retries registration again after encountering a failure condition during last registration. Range: 0 –65535, defaults to 8
5. Sub Retry Intvl: Interval to wait before the phone retries subscriber again after encountering a failure condition during last subscriber. Range: 0 –65535, defaults to 10

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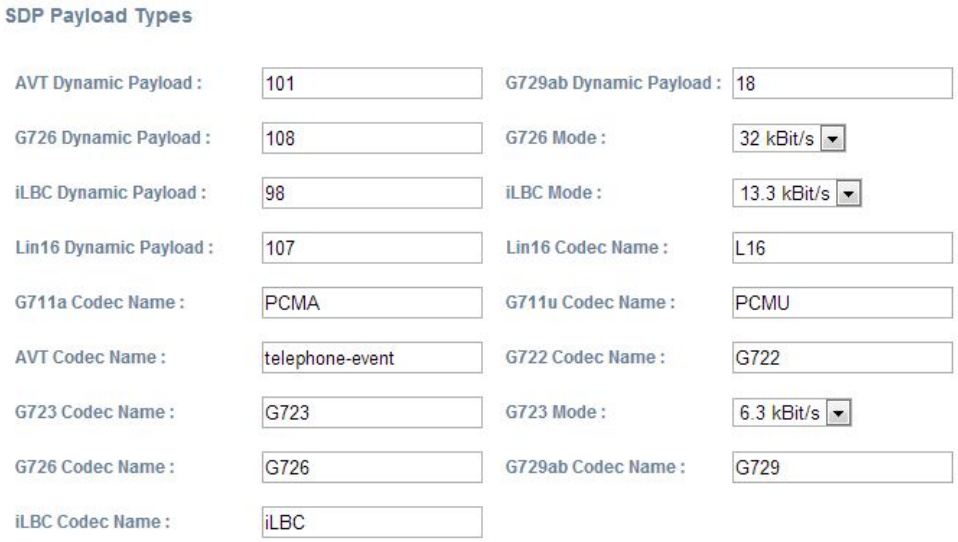
- **RTP Parameters**

1. RTP Port Min: Minimum port number for RTP transmission and reception. Range: 1–65535, defaults to 16384
2. RTP Port Max: Maximum port number for RTP transmission and reception. <RTP Port Max> should be at least 2 larger than <RTP port Min>.Range: 1–65535, defaults to 16482
3. RTP Packet Size(ms): Packet size in milliseconds, which can be 10ms, 20ms, 30ms, 40ms, 60ms

- **SDP Payload Types**



1. AVT Dynamic Payload---RFC2833 payload type. Defaults to 101
2. G729b Dynamic Payload---G729b dynamic payload type. Defaults to 18
3. G726 Dynamic Payload ---G726 dynamic payload type. Defaults to 108
4. G726 Mode ---G726 codec rate, which can be 16 kBit/s,24 kBit/s,32 kBit/s,40 kBit/s. Defaults to 32kBit/s
5. iLBC Dynamic Payload---iLBC dynamic payload type. Defaults to 98
6. iLBC Mode---iLBC codec rate, which can be 13.3kBit/s, 15.2kBit/s. Defaults to 13.3kBit/s
7. Lin16 Dynamic Payload ---Lin16 dynamic payload type, defaults to 102
8. G711a Codec Name---G711a codec name used in SDP, defaults to PCMA
9. G711u Codec Name---G711u codec name used in SDP, defaults to PCMU
10. AVT Codec Name---AVT codec name used in SDP, defaults to telephone-event
11. G722 Codec Name---G722 codec name used in SDP, defaults to G722
12. G723 Codec Name---G723 codec name used in SDP, defaults to G723
13. G723 Mode---G723 codec rate, which can be 6.3kBit/s, 5.3kBit/s, defaults to 5.3kBit/s

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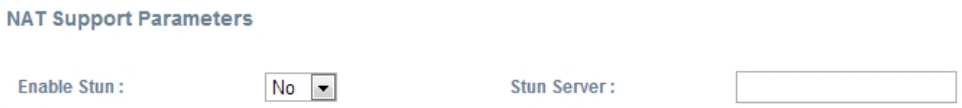
14. G726 Codec Name---G726 codec name used in SDP, defaults to G726
15. G729ab Codec Name---G729ab codec name used in SDP, defaults to G729
16. iLBC Codec Name---iLBC codec name used in SDP, defaults to iLBC

➢ **NAT Support Parameters**

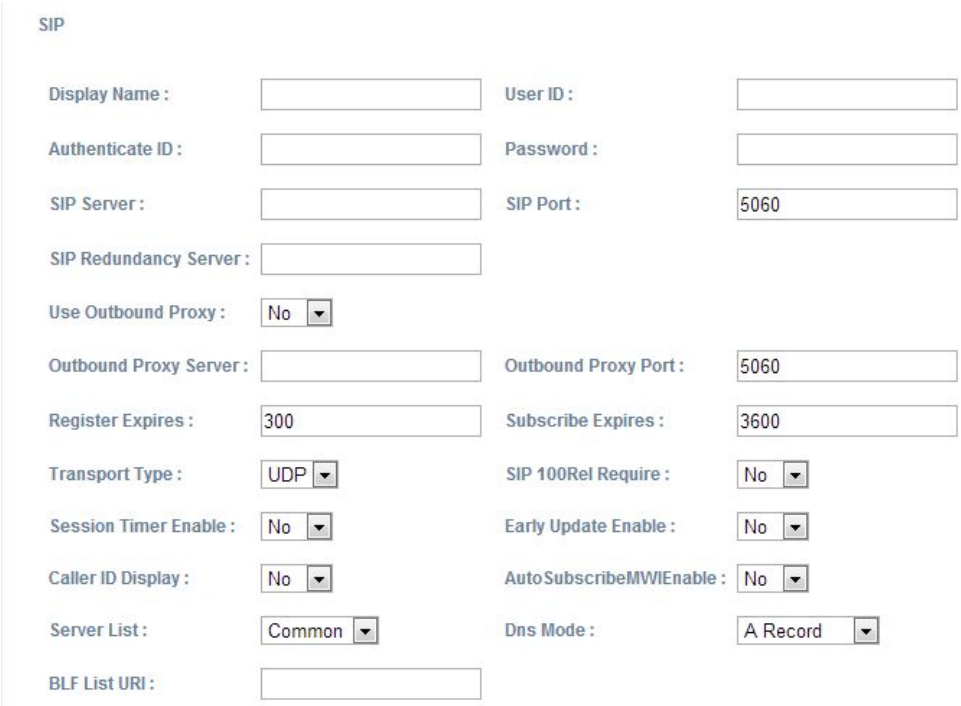


1. Enable Stun: Select "Yes" or "No" to enable or disable using stun to discover NAT mapping.
2. Stun Server: Set stun server, which can be IP address or domain name.

4.4 Account

Rainbow1 has 1 line which is enabled to register by default.

➢ **SIP**



1. Display Name: This name will be displayed on the LCD. It will show the User ID instead if leave Display Name as blank.
2. User ID: Username of sip account.
3. Authenticate ID: Normally is the same as User ID, but blank is acceptable.
4. Password: Password of SIP account.
5. SIP Server: SIP server address, support both IP address and domain name.
6. SIP Port: SIP server port, defaults to 5060.
7. SIP Redundancy Server: SIP redundancy server address.
8. Use Outbound Proxy: Select "Yes" or "No" to enable or disable outbound proxy.
9. Outbound Proxy Server: Set address of Outbound proxy server. All signaling requests will be sent to outbound proxy server firstly.

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10. Outbound Proxy Port: Outbound proxy server port.
11. Register Expires: Register expiration time, defaults to 300 seconds.
12. Subscribe Expires: Subscriber expiration time, defaults to 3600 seconds.
13. Transport Type: UDP/TCP/TLS. Defaults to UDP.
14. SIP 100Rel Require: Select "Yes" or "No" to enable or disable 100Rel. If enabled, 100rel parameters will be added to the SIP request to support PRACK.
15. Session Timer Enable：Select "Yes" or "No" to enable or disable Session Timer.
16. Early Update Enable：Select "Yes" or "No" to enable or disable Early Update.
17. Caller ID Display: Select "Yes" or "No" to enable or disable Caller ID display
18. AutoSubscribeMWIEnable：Select "Yes" or "No" to enable or disable SubscribeMWI
19. Server List：Choose the server type.
20. DNS Mode: Choose the DNS mode
21. BLF List URI: Set BLF list URI when Rainbow1 cooperates with Broadworks (Broadsoft).

- **Codec Configuration**

1. Prefer Codec: Select a preferred codec for all calls. Defaults to G711u. Conversation will be set up by the prefer codec preferentially if this codec also supported by the remote end. However, the actual codec used in a call still depends on the outcome of the codec negotiation protocol.
2. User Prefer Codec Only: Only use the preferred codec for all calls. The call will fail if this codec not supported by the remote end.
3. DTMF Tx Method: Select the method to transmit DTMF signals to the remote end: Inband, RFC2833, SIP INFO. Defaults to RFC2833.

- **Call Feature Setting**

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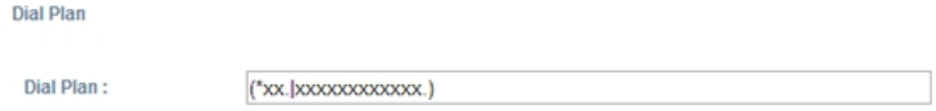
1. Message Waiting: Select "Yes" or "No" to enable or disable indication of new voicemail existed.
2. Voice Mail Number: Set voicemail number.
3. Pickup Service Code: Set Pickup Service Code.
4. UDP Keep Alive Enable: Select "Yes" or "No" to enable or disable UDP keep alive. If enabled, Rainbow1 sends UDP packets periodically to keep the server port alive.
5. UDP Keep Alive Intvl: Set interval to send UDP packets.
6. Default Ring: Set default ring tone.
7. SRTP: Select "Yes" or "No" to enable or disable SRTP (Secure Real Time Control Protocol). This feature will be available only when the server supports SRTP.

- **Dial Plan**



Dial Plan: Configure dial rule for SIP account, please refer to dial plan.

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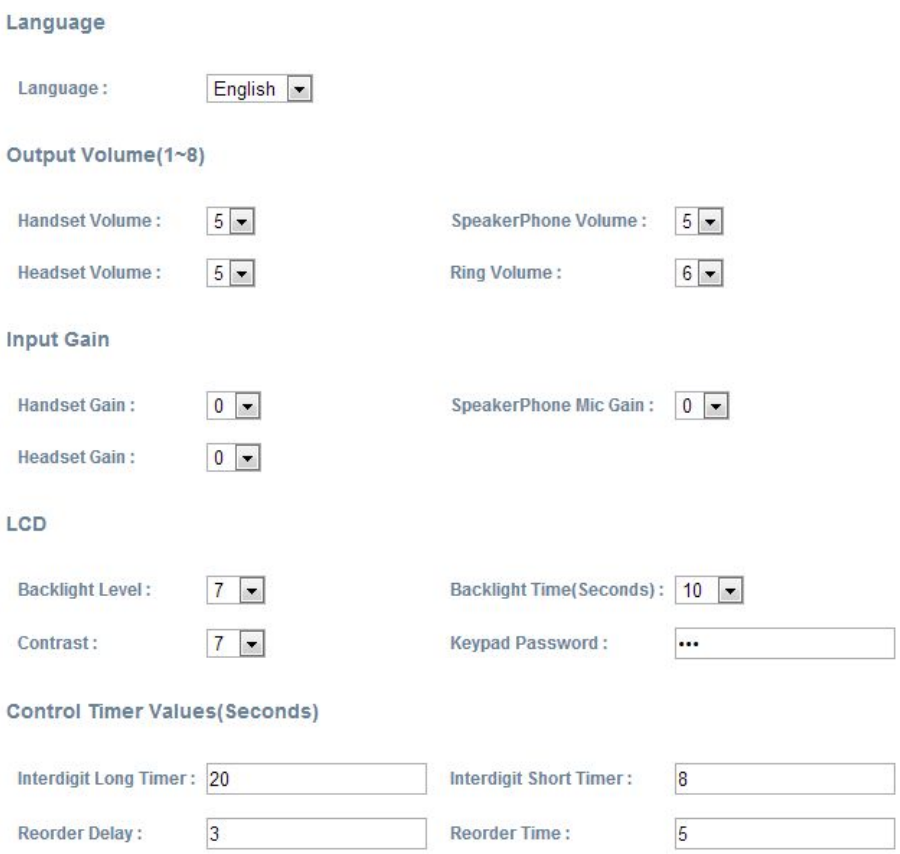
4.5 Phone Setting

4.5.1 Preference



- **Language**

Support customizable multi-language.

- **Output Volume(1~8)**

1. Handset Volume: Specify handset volume grade
2. SpeakerPhone Volume: Specify speaker volume grade
3. Headset Volume: Specify headset volume grade
4. Ring Volume: Specify ring tone volume grade

- **Input Gain**

1. Handset Gain: Specify handset gain, the bigger the gain, the louder the other party heard.
2. SpeakerPhone Mic Volume: Specify speaker gain, the bigger the gain, the louder the other party heard.
3. Headset Volume: Specify headset gain, the bigger the gain, the louder the other party heard.

- **LCD**

1. Backlight Level: select the backlight level
2. Backlight Time(Seconds): select the backlight time
3. Contrast: select the contrast level
4. Keypad Password: set keypad access password

- **Control Timer Values(Seconds)**

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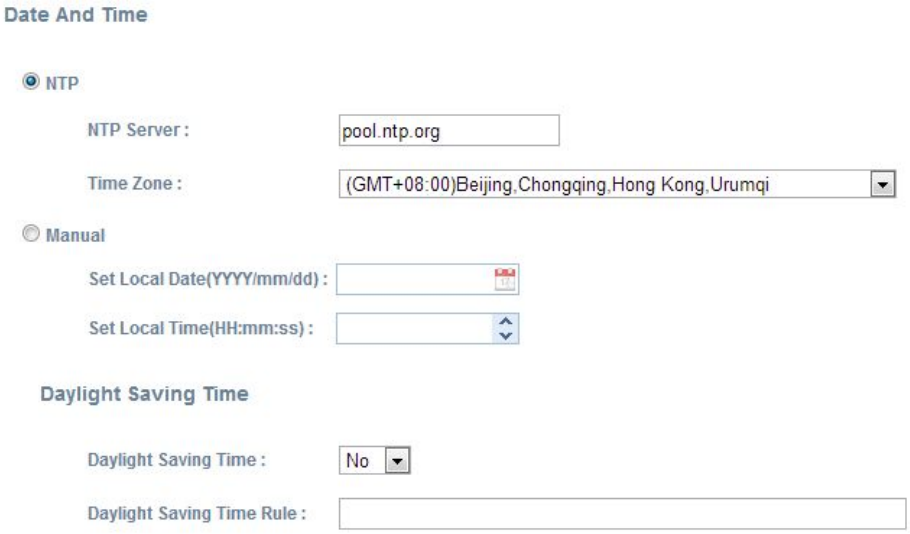
1. Interdigit Long Timer: If the numbers or characters input have not finished and do not full matched the dial plan, it will be not dialed out automatically until time out. Range: 0 – 64 sec
2. Interdigit Short Timer: If the numbers or characters input are full matched the dial plan, it will be dialed out automatically until time out. Range: 0 – 64 sec
3. Reorder Delay: It means the delay between the remote end hangs up and reorder tone is played. "0" means the reorder tone will be played immediately. Range: 0 – 60 sec
4. Reorder Time: Set the duration about how long the page of Calling–end displayed

- **Date And Time**



NTP: Network time protocol

1. NTP Server: Set NTP server address, IP address or domain name are both acceptable.
2. Time Zone: Choose your own time zone.
3. Daylight Saving Time: Select "Yes" or "No" to enable or disable Daylight saving time.
4. Daylight Saving Time Rule: This parameter is a rule with three fields, each field is separated by semicolon; as show below:
   Start=<start-time>; end=<end-time>; save=<save-time>

<start-time> and <end-time> specify the start and end date of daylight saving time, month/date/week and <save-time> is the amount of hour/min/sec to add to the current time during daylight saving period. The <save-time> value can be preceded by a negative (-) sign if subtraction is desired instead of addition.

If <weekday > is 0, it means the date to start or end daylight saving is at exactly the given date. In that case, the <day> value must not be negative. If<weekday >is not zero, then the daylight saving starts or ends on the <weekday >on or after the given date if <day> is positive, or on or before the given date if <day> is negative. If <day> is -1, it means the <weekday> on or before the end-of-the –month (in other words the last occurrence of <weekday> in that month)

Optional values inside [] are assumed to be 0 if they are not specified. Midnight means 0:0:0 of the given date.

E.G. start=4/1/7/8:00:00;end=10/1/7/9:00:00;
save=-1Start time is 1st April, Sunday, 8:00:00 am.
End time is 1st October, Sunday, 9:00:00 am.
The display time will be one hour early than the standard time.

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Manual

1. Set Local Date(YYYY/mm/dd): manually set local date or click to choose local date .Format: year/month/day.
2. Set Local Time(HH:mm:ss) : manually set local time or click to adjust local time. Format: hour/minute/second. e.g. 12:00:00.

4.5.2 Features

- **Speed Dial**

Set speed dial for 2-9 digit key. For example, set Speed Dial 2: 123456789, then if you press 2, the phone number (123456789) will be dialed immediately. The other digit keys 3/4/5.../9 also can be set as speed dial.

- **Call Forward**

1. Always Target: Every incoming call will be forwarded to this target.
2. Busy Target: The incoming call will be forwarded to this target when Rainbow1 is busy.
3. No Answer: The incoming call will be forwarded to this target when there is no answer.
4. After Ring Time(Seconds): After this time out, the incoming call will be forwarded to this target if no answer. Defaults waiting for 5 seconds.
   The priority of Always Target is highest when Always Target, Busy Target and No Answer Target have been set. This means all the incoming call will be forwarded to the Always Target.

- **Call Settings**

1. Do Not Disturb: Select "Yes" or "No" to enable or disable DND (Do Not Disturb).When DND enabled, all the incoming calls will be rejected. At this moment if Always Target or Busy Target

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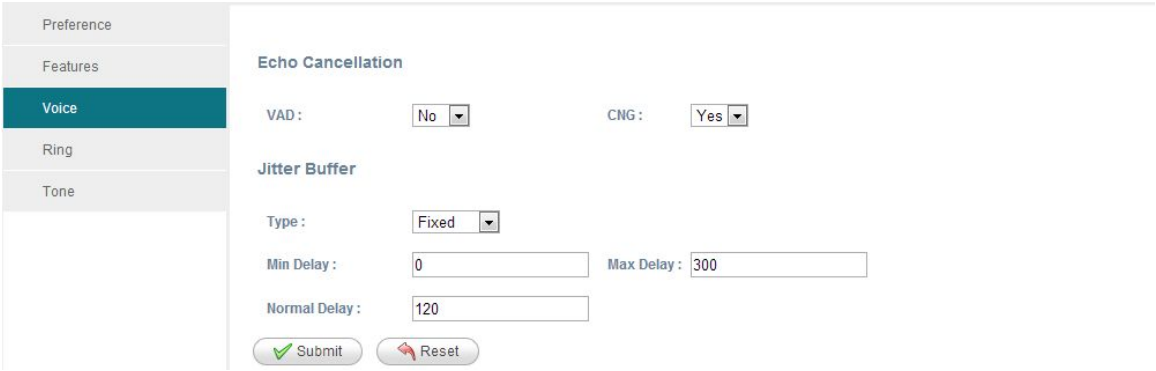
was set, all incoming calls will be forwarded to the targets preferentially.

2. Call Waiting：When Rainbow1 is on calling, and Call Waiting enabled , any new incoming calls will not be rejected but ringback tone can be heard by the remote end. if Call Waiting disabled, only one conversation will be available even multi-accounts have been registered on Rainbow1 .
3. Block Call ID: Select "Yes" here, Caller ID will be blocked, Rainbow1 will call others as anonymous.
4. Block Anc Call: Select "Yes" here, all anonymous call will be blocked, Rainbow1 will reject all anonymous callers.
5. Auto Answer: Select "Yes" here, Rainbow1 will answer all incoming calls by speaker automatically.
6. Send Key: Define the Send key as "#" key or "*"Key. Press "#" Key or "*"Key follow those phone number you want to dial, then number will be dialed out immediately.

4.5.3 Voice



Echo Cancellation

1. VAD：Select "Yes" or "No" to enable or disable VAD (Voice Active Detection). If enable, RTP packets will not be sent when Rainbow1 is mute.
2. CNG：Select "Yes" or "No" to enable or disable CNG (Comfort Noise Generator). If enable, comfortable noise will be sent to the remote end to let it perceive the conversation is still active when Rainbow1 is mute.

Jitter Buffer

Rainbow1 is able to buffer incoming voice packets to minimize out-of-order packet arrival. This process is known as jitter buffer.

1. Type: Choose type of jitter buffer. When choose Fixed, the size of jitter buffer is fixed. When choose Adaptive, the size of jitter buffer is the sum of Minimum Delay and the size of RTP packets.
2. Min Delay: The minimum delay of the jitter buffer.
3. Max Delay: The maximum delay of the jitter buffer.
4. Normal Delay: This is used to set fixed jitter buffer which should be between Min Delay and Max Delay.

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4.5.4 Ring

Administer can upload 2 user define ring. The ring file should be wav (8k, 8bit, u-law) and no larger than 200 KBytes.

4.5.5 Tone

Set the ToneScript for each tone.
For example, MWI Dial Tone: 350@-19,440@-19;2(.1/,1/1=2);10(*/0/1+2)
Frequency1, Frequency2 ;Cadence Section1;Cadence Section2
350@-19: Frequency1 is 350HZ at -19dBm
440@-19: Frequency2 is 440HZ at -19dBm
2(.1/,1/1=2): Cadence Section length is 2s, 0.1s on, 0.1s off, with frequencies 1 and 2
10(*/0/1+2): Cadence Section length is 10s, always on(* means always, 0 means never), with frequencies 1 and 2.

1. Country stands: Select a country standard tone as default.
2. Dial Tone：The prompting audio to remind user to enter a phone number for dialing
3. Outside Dial Tone：The prompting audio to remind user to enter an external phone number (versus an internal extension). This is triggered by a "," character encountered in the dial plan.
4. Busy Tone：The promoting audio when a 486 RSC is received for an outbound call.
5. Reorder Tone：The promoting audio when an outbound call failed or after the remote end hangs up an established call.
6. Off Hook Warning Tone: The promoting audio when user does not put the handset on the cradle properly.
7. Ring Back Tone：The promoting audio when the remote end is ringing.
8. MWI Dial Tone：The promoting audio when an unread voicemails existed.

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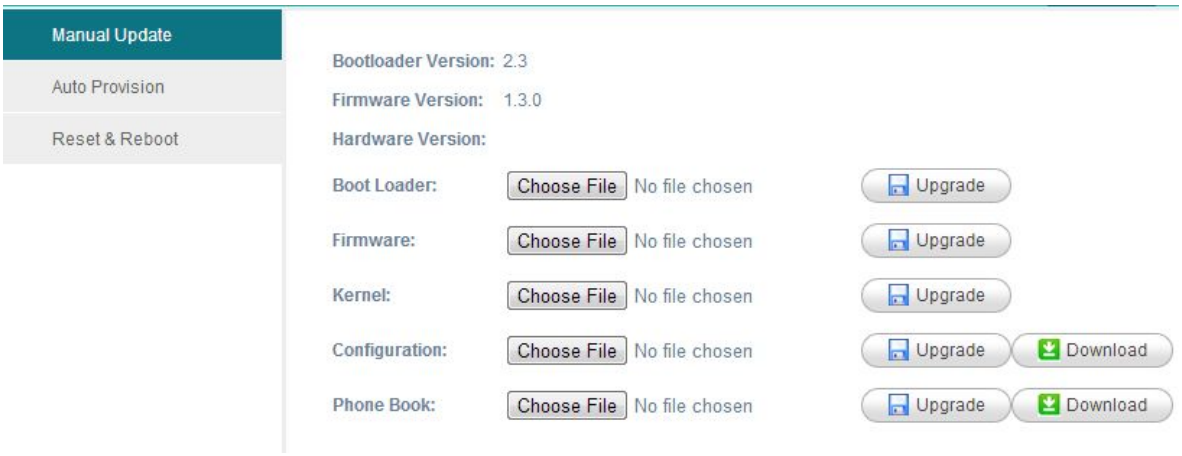
9. Holding Tone：Indicate the local end that the current calling is hold by the remote end.
10. Conference Tone：The promoting audio to all parties when a 3 way conference is in progress.

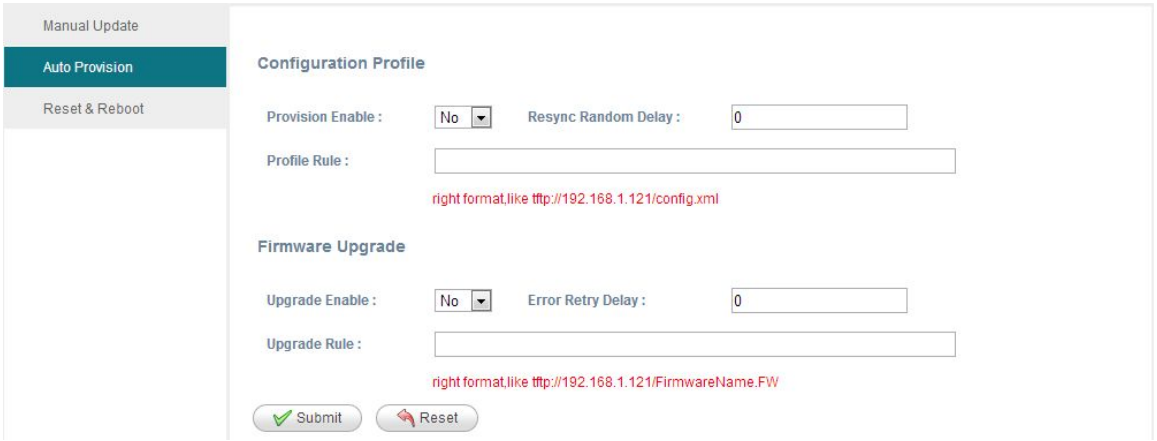
4.6 Update

4.6.1 Manual Update


Manual Update
Auto Provision
Reset & Reboot
Bootloader Version: 2.3
Firmware Version: 1.3.0
Hardware Version:
Boot Loader: Choose File No file chosen Upgrade
Firmware: Choose File No file chosen Upgrade
Kernel: Choose File No file chosen Upgrade
Configuration: Choose File No file chosen Upgrade Download
Phone Book: Choose File No file chosen Upgrade Download

1. Boot Loader: Download Boot file from ATCOM's webpage, then select the file from your PC to upgrade.
2. Firmware: Download firmware from ATCOM's webpage, then select firmware from your PC to upgrade.
3. Kernel: Download kernel from ATCOM’s webpage, then select kernel from your PC to upgrade.
4. Configuration: Upload and download config.xml file.
5. Phone Book: Upload and download phonebook.xml file

4.6.2 Update/Auto Provision



Configuration Profile

1. Provision Enable： Select “Yes” or “No” to enable or disable auto provision
2. Resync Random Delay: Set a random delay to upgrade configuration files.
3. Profile Rule: FTP/TFTP/HTTP protocol supported

Firmware Upgrade

1. Update Enable: Select “Yes” or “No” to enable or disable firmware update
2. Error Retry Delay: Set interval to retry update after encountering update failed

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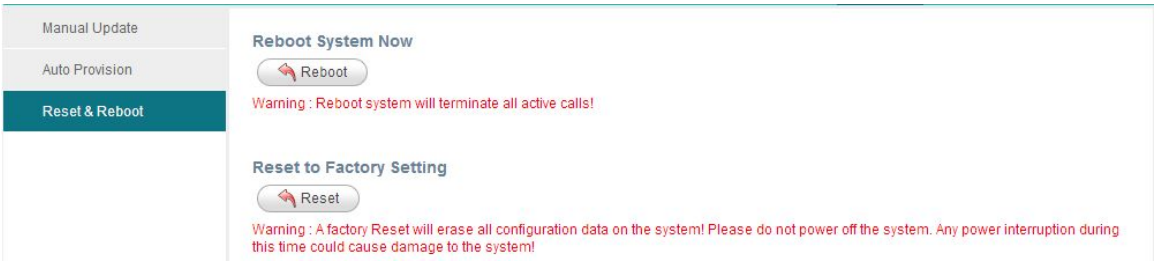
3. Upgrade Rule: The same as Profile Rule, the suffix of the firmware name should be '.fw' or '.FW' and the firmware name includes blank character or Chinese characters are not permitted.

4.6.3 Reset & Reboot


Manual Update
Auto Provision
Reset & Reboot
Reboot System Now
Reboot
Warning : Reboot system will terminate all active calls!
Reset to Factory Setting
Reset
Warning : A factory Reset will erase all configuration data on the system! Please do not power off the system. Any power interruption during this time could cause damage to the system!

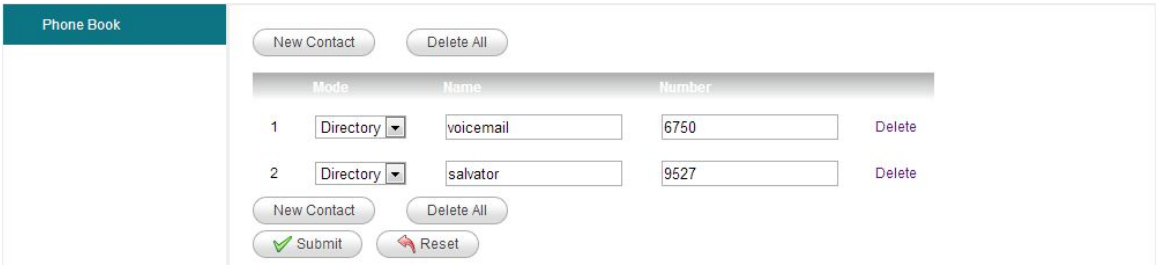
1. Reboot

Reboot will terminate all active calls, and restart the phone in a several seconds.

2. Reset

Click the "Reset" button will set Rainbow1 to factory default, please backup the config.xml, phonebook.xml and attendant_keypad.xml before reset.

4.7 Phone Book



1. New Contact: Add a new phonebook record.
2. Delete All: Delete all the phonebook records.
3. Mode: Directory or Blacklist. All calls from Blacklist will be rejected.
4. Name: Specify a name for each phone number. This name will be displayed on LCD when call to this number or call from this number.
5. Number: Phone number.
6. Submit: Submit to apply the change.
7. Reset: Cancel the inputting.
8. Delete: Delete this record.

Notice：The maximum of records in the phone book is 100.

4.8 Call Log

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1. Redial: Record of dialed list, maximum 100 records.
2. Received: Record of received list, maximum 100 records.
3. Missed: Record of Missed list, maximum 100 records.

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5. FAQ & Trouble Shooting

Frequently Asked Questions

5.1 How to make a factory reset

There are three ways to make a factory reset:

1. Factory reset from keyboard, steps are:

a) Press and input password 123.

b) Find and select '12 Factory Reset'.

c) Press 'OK' key, then the phone will be restarted automatically and factory reset.

2. Factory reset through web, please refer to Reset & Reboot;

3. Factory reset during rebooting, steps are:

a) Press '*' and '#' key immediately after plugging in the power adapter.

b) Wait for about 5 seconds, Rainbow1 will be reset to factory default setting after rebooted.

5.2 Upgrade firmware under safe mode

If the phone could not start up normally, please upgrade firmware under safe mode. Steps are:

1. Build a TFTP server and set its IP address as 192.168.1.200.
2. Copy the firmware to the root directory of the TFTP server and modify the file name as Rainbow1.FW.
3. Make sure the TFTP server and the phone are connecting with the same switch and reachable to each other;
4. Keep pressing '#' key and start up the phone until the LCD displays 'Upgrading…';
5. The phone will download the firmware from the TFTP server;
6. After the download is done, the phone will upgrade automatically;
7. After the update is finished, the phone will start up with the factory configuration.

5.3 How to make direct IP call

When hook off/ pressing speaker or headset key,

1. Press soft key 'Num' until it switch to 'IP'.
2. Using '*' key to input '.'. If user wants to dial 192.168.1.100, then press 192*168*1*100.
3. After inputting the IP address, press dial key to dial it out.

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6. Trouble Shooting

6.1 The phone can't register successfully

1. Check the IP address, and if the mode of WAN port is DHCP, please make sure the DHCP server is in service.
2. Check the gateway.
3. Check the DNS.
4. Make sure the information of the account is consistent with which offered by the service supplier.
5. Make sure the SIP server is on.
6. Check the port of the SIP server whose default value is 5060.

6.2 The phone can't obtain IP address

1. Make sure the cable has been connected to the LAN port of the phone.
2. Make sure the cable and the switch's port that connected with the cable is available.
3. Make sure the DHCP server is on and there are some assignable IP addresses in the address pool.
4. Try to change the LAN port mode as Static.

6.3 Only one part can hear the voice during the call

1. Make an IP dial-up call to make sure the telephone receiver and microphone are normal.
2. Enable STUN on web page.
3. Set STUN server as stun.sipgate.com.
4. Click 'submit' and wait for the phone to restart.
5. Try to make calls again.

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7. Abbreviations

DND ： Do Not Disturb
CFWD ： Call Forward
Bxfer ： Blind Transfer
Conf ： Conference
Num ： Number
DelChr ： Delete Char
Y/N ： Yes/No
SIP：Session Initiate Protocol
RTP：Real-time Transport Protocol
SDP：Session Description Protocol
VPN：Virtual Private Network
VLAN：Virtual Local Area Network
QoS：Quality of Service
Syslog ： System log
UDP：User Data Protocol
TCP：Transmission Control Protocol
TLS：Transport Layer Security Protocol
BLF：Busy Lamp Field
DNS：Domain Name System
SRTP：Secure Real-time Transport Protocol
NTP：Network Time Protocol
VAD：Voice Activity Detection
CNG：Comfort Noise Generator

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